DETAILED ACTION

1. This action is response to application number 10590937, amendment and arguments dated on 03/02/2010.

Response to Arguments

2. Applicant's arguments filed 03/02/2010 have been fully considered but they are not persuasive.

Applicant argues in page 3 of arguments that "Vimpari teaches away from the "problematic method". and further argue that "Teaching away was one indicia of nonobviousness recognized explicitly by the Supreme Court in its *KSR* decision. Clearly, paragraphs [0008], [0017], and [0018] in Vimpari discourage a person of ordinary skill from increasing the size of packets sent over the link when the rate of packet loss is unacceptably high or decreasing the size of packets sent over the link when the rate of packet loss is within acceptable limits according to the claims".

Vimpari in ¶0036 discloses "There may advantageously be one threshold value per each RTP packet length so that <u>the RTP packet length can be changed up or down</u> by one basic packet e.g. <u>according to the FER (Frame error rate) measured</u>", ¶0038 discloses "<u>the number of basic packets to be included in the RTP packet can be either increased or decreased</u>. <u>If the frame error rate indicates an increase in the number of errors</u>". The method of processing variable length packet (title) by Vampari is able to increase or decrease the packet length based on the link measured frame error rate. Therefore it is a design choice to increase and decrease RTP packet length when FER changed (increased or decreased). In other words, the disclosed method by Vampari

can be configured to increase packet length when FER increased, or increase packet length when FER decreased. Furthermore, Vimpari in ¶0008 describes the method that it is well known in the art, increasing packet length sent over the link when the frame error rate (FER) is high, in other words, decreasing the packet length when the link condition improves.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention was made to increase the packet size sent over the link when the frame error rate (FER) is higher than a threshold in order to decrease the average proportion of header data per one packet and reducing the packet header overhead, as taught by Vimpari.

Applicant argues in pages 4 that “Consequently, Vimpari is not concerned with reducing bandwidth usage over the radio interface, but rather with reducing delays for a particular call using that radio interface”.

Vimpari in ¶0003 discloses “Large headers in each packet transferred slow down packet processing and eat up transfer capacity”, to solve the issue of slowing down (delay) and eating up capacity (bandwidth) by overhead bits of packets (headers) in a cellular network link, Vimpari discloses increasing length of packet based on frame error rate. Vimpari further discloses in ¶0008 that "The capacity-consuming (bandwidth consuming) effect of the header can be reduced by including in one packet more data blocks, e.g. one packet could contain three such data blocks". Therefore, Vimpari discloses a method to use more efficiently the bandwidth (capacity) of the link (link,

radio interface) and reducing delay by decreasing the average proportion of header data to payload data, by adjusting the RTP packet size based on the link frame error rate.

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

3. Claims 8-14 is rejected under 35 U.S.C. 103(a) as being unpatentable over by Vimpari Markku (US. 2003/0117972).

Claim 8, Vimpari discloses a method (abstract, page 1) of optimizing the bandwidth usage (¶0015) on a real Time Protocol (RTP) managed link transporting media (communication connection) from a Media Resource Function node (Fig. 1, el. 14, converter node) of a cellular telecommunications network (Fig. 1, el. 11) to User Equipment (Fig. 1, el. 12b, terminal), the method comprising:

at Media Resource Function node (Fig. 1, el. 14, converter node), monitoring (¶0011) the rate of packet loss (frame or packet loss; ¶0035) of the link to determine (¶0036) whether the rate of packet loss is unacceptably high or within acceptable limits (measuring quality parameters of communication connection; ¶0011); and

as a result of said monitoring (¶0011), adapting the sending rate from at Media Resource Function node (Fig. 1, el. 14, converter node) over the link (communication connection, Fig. 1, els. 13a and 13b) to the user Equipment by re-packetising media (increase or decrease the number of data blocks in a single RTP packet; Fig. 2, el 24; ¶0038), received at the Media Resource Function node (converter node, Fig. 1, el. 14) from third party nodes (Fig. 1, el. 12a), to either increase the size of packets sent over the link when the rate of packet loss is unacceptably high (Vimpari chooses to decreasing size of transmission packet in case of excessive transmission packet loss; ¶0038), thereby reducing packet header overhead and reducing bandwidth usage on the link (Vimpari selection of smaller size in configuration of the packet in case of higher packet loss is base on fact that losing bigger size packet and retransmission of those packet would lead to less effective usage of bandwidth; ¶0008); or to decrease the size of packets sent over the link when the rate of packet loss is within acceptable limits (Fig. 2, el. 22, ¶0038), thereby reducing the transmission delay over the link (Vampari selection of decreasing size of packet when frame loss rate exceeds a threshold, and increasing the packet size when frame loss rate is within acceptable limits (threshold) is a matter of selecting an option of available configuration. Vampari in ¶0008 discloses a configuration which intend to increase frame (packet) size by adding more packet to the frame when the rate of frame loss (packet loss) is high (for example, a link with a packet loss (frame

loss) of 2% when the frame size increased by 3 times, would lead to 6% of packet loss); ¶0017; ¶0038; Fig. 2).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention was made to increase the packet size sent over the link when the frame error rate (FER) is higher than a threshold in order to decrease the average proportion of header data per one packet and reducing the packet header overhead, as taught by Vimpari.

Claim 9, Vimpari further discloses wherein the step of monitoring (measuring) the rate of packet loss (frame or RTP packet loss) of the link (communication connection, Fig. 1, els. 13a and 13b) comprises sampling (¶0035; Fig. 2, e. 22).

Claim 10, Vimpari further discloses wherein said step of adapting the sending rate is carried out dynamically in response to the monitored rate of packet loss (Fig. 2; Vampari describes after sending the repacketised RTP packet, the device is ready to receive or send the next RTP packet using the RTP packet length adaptor according the step 24; ¶0039).

Claim 11, Vimpari further discloses wherein, in the event that media is to be repacketised (¶0017) at the Media Resource Function node (Fig. 1, el. 14, converter node), received media is stored at the Media Resource Function (Fig.

1, el. 14, converter) in a buffer until such time as sufficient media has been received to construct a packet of the necessary size (in ¶0042, Vimpari describes control unit of converter (Media Resource Function) that disassembles the RTP packets into basic packets if long RTP packet received or combine several basic packets into one RTP Packet for transmission if the frame error rate measurement is in acceptable range (less than threshold). The converter as described must buffer the basic packets after disassembling or before combing them to a larger size RTP packet for transmission when frame error rate allows; ¶0042).

Claim 12, Vimpari further discloses wherein said third party nodes are peer User Equipment (UEs) (Fig. 1, el. 12a; ¶0017).

Claim 13, Vimpari discloses a Media Resource Function node (Fig. 1, el. 14, converter; ¶0027) for use in a cellular telecommunications network (Fig. 1, el. 11), the node handling media sent between itself and user equipment (Fig. 1, el. 12b, terminal), over a Real-Time Protocol managed (RTP) link (communication connection, Fig. 1, els. 13a and 13b), the node comprising:

means for monitoring (¶0011) the rate of packet loss (frame or packet loss; ¶0035) of the downlink to the User Equipment (measuring quality parameters of communication connection 13b and 13a in Fig. 1; ¶0011) to

determine (¶0036) whether the rate of packet loss is unacceptably high or within acceptable limits (Fig. 2, el.22; ¶0035; and

means for adapting (¶0017; ¶0027), based upon the monitored properties (¶0035), the sending rate over the link (communication connection, Fig. 1, els. 13a and 13b) by re-packetising media received from third party nodes (Fig. 1, el. 12a), to increase the size of packets sent over said downlink when the rate of packet loss is unacceptably high (Vimpari chooses to decreasing size of transmission packet in case of excessive transmission packet loss; ¶0038), thereby reducing packet header overhead and reducing bandwidth usage on the link (Vimpari selection of smaller size in configuration of the packet in case of higher packet loss is base on fact that losing bigger size packet and retransmission of those packet would lead to less effective usage of bandwidth; ¶0008); or to decrease the size of packets sent over the link when the rate of packet loss is within acceptable limits (Fig. 2, el. 22, ¶0038), thereby reducing the transmission delay over the link (Vampari selection of decreasing size of packet when frame loss rate exceeds a threshold, and increasing the packet size when frame loss rate is within acceptable limits (threshold) is a matter of selecting an option among available configuration options for the transmission link. Selecting different options of a configuration option does not constitute a new invention; ¶0017; ¶0038; Fig. 2).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention was made to increase the packet size sent over the link

when the frame error rate (FER) is higher than a threshold in order to decrease the average proportion of header data per one packet and reducing the packet header overhead, as taught by Vimpari.

Claim 14, Vimpari discloses a media resource function node (Fig. 1, el. 14, converter node; ¶0027) for use in a cellular telecommunications network (Fig. 1; title), the media resource function node (Fig. 1, el. 14) handling media sent between the media resource function node (Fig. 1, el. 14) and user equipment (Fig. 1, el. 12b) over a real-time protocol managed link (Fig. 1,els. 13b and 13a; ¶0001), the media resource function node (Fig. 1, el. 14) comprising control circuitry configured to:

monitor the rate of packet loss of a real-time protocol (¶0011; ¶0036; ¶0035) managed downlink to the user equipment (Fig. 1, el. 12b; abstract) to determine whether a rate of packet loss (Fig. 2, el. 22) for the real-time protocol (real-time protocol, RTP) managed downlink is unacceptably high or within acceptable limits (packet loss (error) rate within or exceed a predetermined limit; ¶0031; ¶0036); and

adapt (¶0017; ¶0027), based upon the monitored properties (¶0011; ¶0036; ¶0035), the sending rate over the real-time protocol managed downlink by re-packetizing media received from third party nodes (Fig. 1, el. 12a) in order to increase the size of packets sent over the real-time protocol managed (RTP) downlink (¶0027; ¶0031) when the rate of packet loss is unacceptably high

(exceed the predetermined threshold; ¶0031; ¶0036) to reduce packet header overhead and reducing bandwidth usage (¶0017; ¶0019; ¶0002-¶0003; ¶0006; ¶0008) on the real-time protocol managed downlink (RTP) or to decrease the size of packets (¶0036) sent over the real-time protocol managed downlink when the rate of packet loss is within acceptable limits (¶0036; ¶0038) to reduce the transmission delay (¶0003) over the real-time protocol (RTP) managed downlink (Vampari selection of decreasing size of packet when frame loss rate exceeds a threshold, and increasing the packet size when frame loss rate is within acceptable limits (threshold) is a matter of selecting an option and configurable (design choice). Vampari in ¶0008 discloses a configuration similar to present application, which increases frame (packet) size by adding more packet to the frame when the rate of frame loss (packet loss) is high (for example, a link with a packet loss (frame loss) of 2% when the frame size increased by 3 that result to 6% of packet loss); ¶¶0017; ¶0038; Fig. 2).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention was made to increase the packet size sent over the link when the frame error rate (FER) is higher than a threshold in order to decrease the average proportion of header data per one packet and reducing the packet header overhead, as taught by Vimpari.

4. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

The reference Phillips et al. (US 5,490,168) US published date 06 Feb 1996, discloses a method and communication system provides automatic optimization by adjusting the encoder of the transmitter to use a long packet length during low error counts and a short packet length during high error count.

The reference Pazhyannur et al. (US 2003/0161326) US filling date 25 Feb 2002, discloses a method and apparatus to monitor bit error rate of the transmission media and change dynamically the size of transmission packets for optimal frame size.

The reference Dzung Dacfey (EP 1120932 A1) published date 1 Aug 2001 discloses a method in data transmission to use variable packet length base on packet error rate and determines the optimal data length for the transmission.

Conclusion

THIS ACTION IS MADE FINAL. Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to KOUROUSH MOHEBBI whose telephone number is (571)270-7908. The examiner can normally be reached on Monday to Thursday, 7:30AM-5:00PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Chi Pham can be reached on 571-272-3179. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/K. M./
Examiner, Art Unit 2471

/Chi H Pham/
Supervisory Patent Examiner, Art Unit 2471